

5-2014

Transmission of Stereo Audio Signals with Lasers

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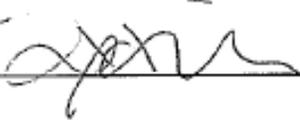
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Thesis Advisor:



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Thesis Committee:

Transmission of Stereo Audio Signals with Lasers

An Undergraduate Honors College Thesis

in the

Department of Electrical Engineering
College of Engineering
University of Arkansas
Fayetteville, AR

by

William Austin Curbow

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I. ABSTRACT

An alternative method for transmitting audio signals via the use of laser technology was presented here. The primary focus of this document is on the audio amplifier subsystem including the design, simulation, and performance results. Simulations using the PSPICE computer simulation software were utilized to investigate the theoretical designs of the audio amplifier subsystem. The audio amplifier was examined in three junctures including the noise-canceling input stage, the high voltage gain cell, and the low impedance driver. Through the design procedure using theoretical calculations, simulations using PSPICE computer software, and examination of the finished product with an oscilloscope, the audio amplifier was greatly investigated and verified for desired functionality. In addition, the pulse-width modulation technique was examined and implemented with precision 555 timers in order to transmit audio signals via laser.

II. INTRODUCTION

In the modern world, audio is disseminated in a myriad of simple and complex enterprises. Examples include, but are not limited to, the basic transmission of analog signals through a copper wire, the more common transfer of FM or AM signals through the air, and a more innovative transfer of signals in a digital format realized by light through a fiber optic cable. This document presents an alternative method to transmitting a stereo audio signal: through lasers with pulse-width modulation. Pulse-width modulation (PWM) is a modulation technique comparable to the common methods of modulation such as frequency modulation (FM) or amplitude modulation (AM) and will be explained in greater depth later in this document. The transmission of audio via laser is not currently economically sensible as the other methods of transmission listed above are more affordable and offer more reliable modes of operation. The main drawback of the laser transmission method presented here is the requirement for the laser's beam to be directly aimed at the receiver, and any slight misalignment at the transmitting side can cause the audio to be interrupted. However, lasers are still a verifiable method of audio transmission, and despite their drawbacks, can produce excellent quality sound reproduction.

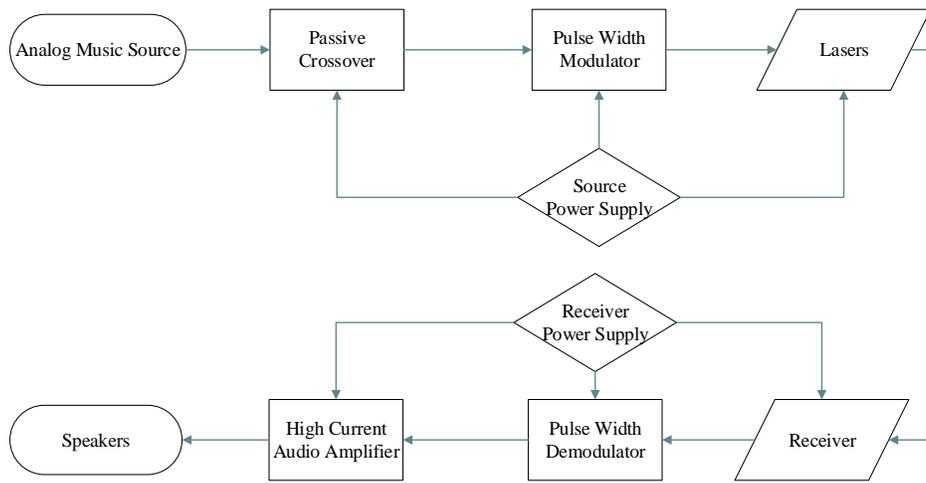


Figure 1. Flow Diagram of Laser Transmission System

Figure 1 on the previous page illustrates the subsystems of the laser transmission system from the origin at the analog audio source to the completion and realization of music at the speakers. The main subsystems designed in this project include the crossover, the pulse-width modulator, the source and receiver power supplies, the pulse-width demodulator, and the audio amplifier. This document will focus primarily on the design, the analysis, and the testing of the audio amplifier subsystem.

III. THEORETICAL BACKGROUND

The pulse-width modulation technique is a common method of modulation often used in voltage source inverters for variable frequency motor drives, in maximum power point trackers for photovoltaic battery chargers, and most notably in closed-loop DC-DC converters. PWM has recently made its way into the audio world with the improvement of the semiconductor devices used to implement the modulation such as the MOSFET and BJT. Accurate reproduction of sound requires an extremely high switching rate with high quality systems switching in the hundreds of megahertz [1].

PWM requires two signals: a high frequency triangular carrier signal and the desired audio or sinusoidal signal. The concept of PWM is straight forward and is illustrated in Figure 2 on the next page [2]. The top plot of Figure 2 shows the high frequency triangular waveform and the much lower frequency sinusoidal signal. The carrier frequency must be a steady triangular signal in order to accurately reconstruct the audio signal after rectifying the PWM signal. A comparator or operational amplifier is used to compare the magnitudes of the carrier signal and the audio signal. If the magnitude of the sinusoidal audio signal is less than the magnitude of the triangular carrier signal, then the PWM output is low or 0 V.

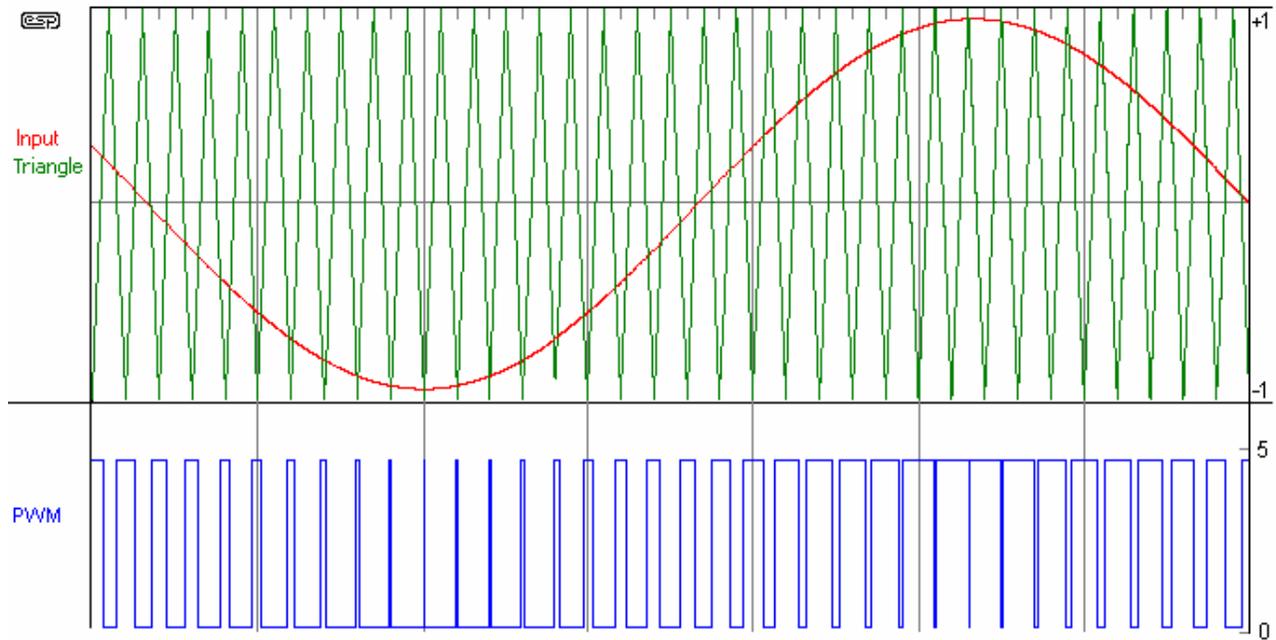


Figure 2. Basic PWM Waveform from Single Frequency Sinusoid

On the other hand, the opposite is also true. If the magnitude of the sinusoidal audio signal is greater than the magnitude of the triangular carrier signal, then the PWM output is high or 5 V as shown in Figure 2. This previously analog signal can now be accurately transmitted with zeroes and ones or low and high voltages with any digital transmission process.

With the original analog audio signal in a digital form, it may be transmitted with a laser. This process requires a fast-switching BJT or MOSFET driver that is capable of switching at the frequency of the carrier signal. The laser driver and laser beam pulse for different lengths of time as shown in the bottom plot of Figure 2. The following step consists of the acknowledgement of the signal by a photovoltaic receiver. The receiver must also be able to emanate a voltage proportional to that of the laser's beam as quickly as the carrier frequency is switching. The design of this subsystem will be discussed in the following section, Design Procedure.

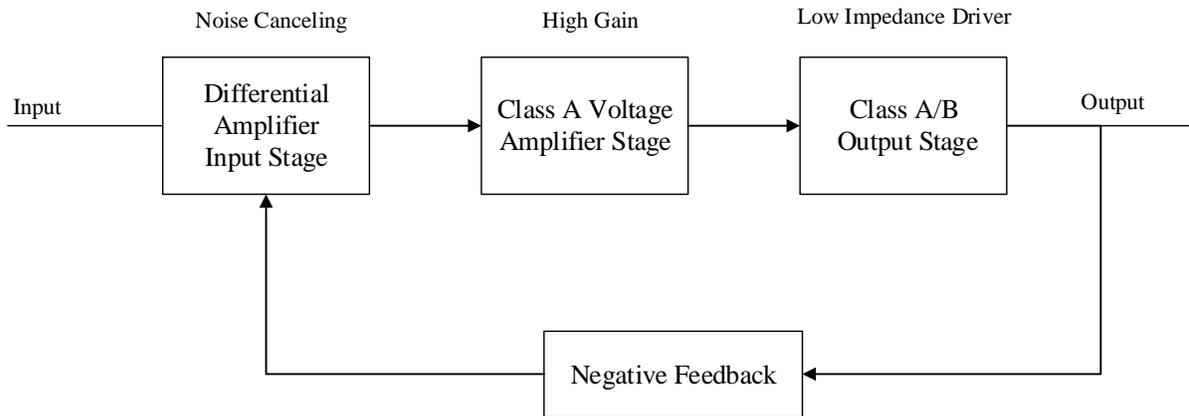


Figure 3. Audio Amplifier Flow Diagram

Here, the audio amplifier will be discussed in great detail focusing on the circuit building blocks that make up the subsystem. There are three main junctures to the audio power amplifier: the noise-canceling input stage, the high voltage gain cell, and the low impedance output driver. Figure 3 illustrates the flow diagram of the audio amplifier subsystem. First, the noise-canceling input stage is implemented with a circuit topology called a differential amplifier. According to *Microelectronic Circuits* by Sedra and Smith, “The differential amplifier is the most widely used building block in analog integrated-circuit design. For instance, the input stage of every op amp is a differential amplifier” [3]. The main difference between differential and single-ended amplifiers is that the differential amplifier is much less susceptible to noise, which makes it the perfect input stage to an audio amplifier for reducing the total harmonic distortion. The noise-proof characteristic of the differential amplifier is achieved through the rejection of common-mode signals while amplifying the differential signal from the input. Figure 4 shows a BJT differential amplifier circuit topology [1]. In a basic audio power amplifier, the analog input is fed to the base of Q1, the input signal ‘A’ position in Figure 4, while the output is fed back to the base of Q2, the input signal B position in Figure 4, at a small percentage of its magnitude.

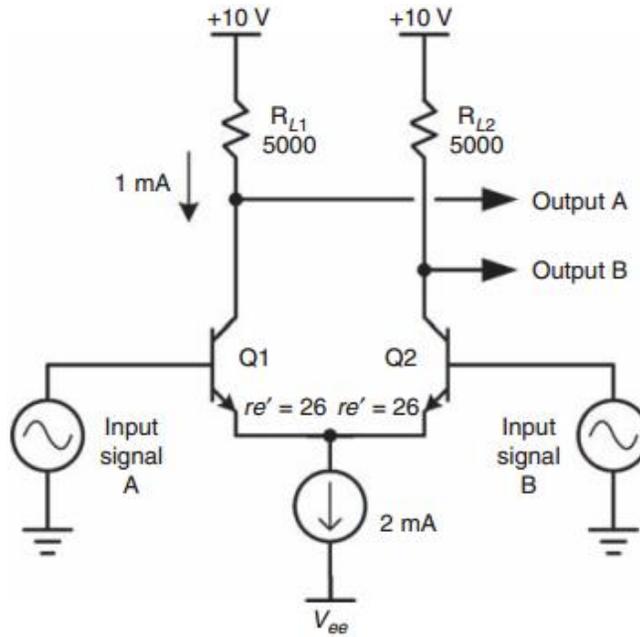


Figure 4. BJT Differential Amplifier Circuit Topology

Next, the main gain stage of the audio power amplifier is implemented using a common-emitter topology. This topology is referred to as the basic gain cell of the amplifier as it is simple, requiring only one transistor, and it is capable of producing a very large voltage gain. According to Bob Cordell in *Designing Audio Power Amplifiers*, “We thus have, for the CE stage, the approximation:

$$A_v = \frac{R_L}{(r_e + R_e)}$$

where R_L is the net collector load resistance and R_e is the external emitter resistance” [1]. It can easily be seen that a high load resistance gives way to a high voltage gain. Because of this relationship, the basic gain cell cannot be used to drive low impedance loads such as a speaker. This low impedance driver is the last stage in the audio amplifier and is directly fed from the output of the common emitter stage.

The function of the output stage of the audio power amplifier is to offer a low output resistance, which is used to drive low impedance loads such as speakers without sacrificing any

gain. One of the most important characteristics of the output stage is the efficiency, which can range from the extremely low efficiency of the class A output stage, which is typically 25%, to the nearly ideal output stage of the class D amplifier, which can achieve an efficiency greater than 95%. The audio power amplifier in this project utilizes the class AB output stage topology shown in Figure 5 below. This output stage is also known as the push-pull output stage as the NPN transistor, Q_P , is used to ‘push’ or source current into the output while the PNP transistor, Q_N , is used to ‘pull’ or sink current from the load. The input shown in Figure 5 comes from the basic gain cell or the common-emitter voltage gain stage as discussed in the previous paragraphs. The gain in the output stage is ideally 1 V/V, which essentially passes the gain derived from the previous stage to the speakers.

The difference between the class AB output stage and the class B output stage lies in the bias on the bases of the two transistors. The class B output stage does not utilize a bias on the bases of its transistors and therefore has a problem with crossover distortion, which consists of a dead band at small amplitudes where both transistors are off. The class AB amplifier is capable of completely removing the crossover distortion and also has a relatively high efficiency; therefore, it is much more widely used as an amplifier’s output than the class A or class B arrangements.

All that remains for the output stage is acquiring the necessary bias for the push-pull transistors. This is most efficiently done with a circuit known as a V_{BE} multiplier. As its name suggests, the V_{BE} multiplier is used to obtain a bias voltage equal to some multiple of the base-emitter voltage. Figure 5 shows a V_{BE} multiplier consisting of R_1 , R_2 , and Q_1 in a class AB output stage.

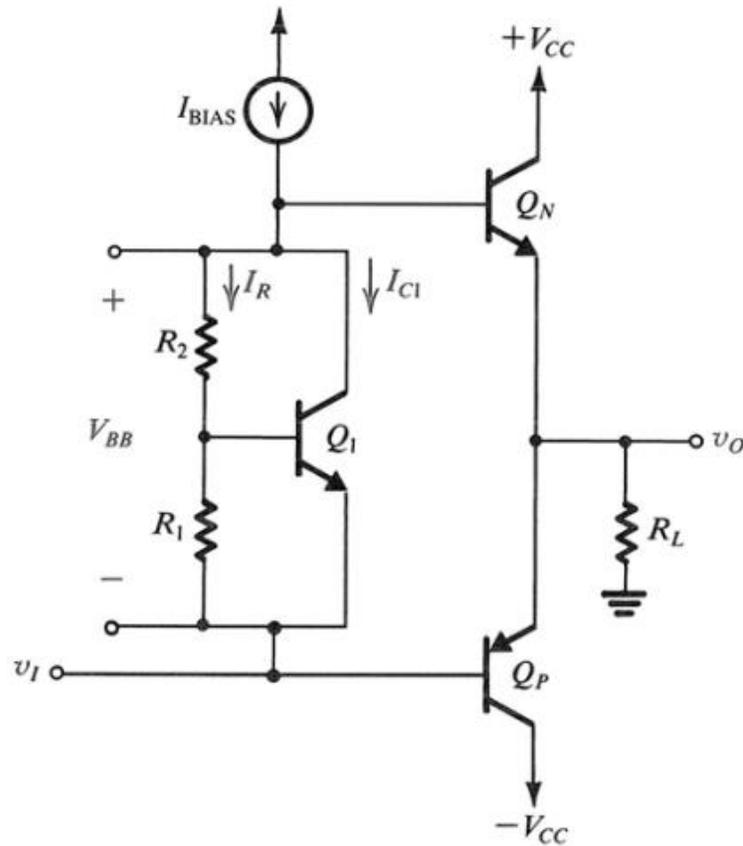


Figure 5. Class AB Output Stage with V_{BE} Multiplier for Bias

Microelectronic Circuits explains the relationship between the bias voltage achieved with the V_{BE} multiplier and the values of R_1 and R_2 [3].

$$V_{BB} = V_{BE1} \left(1 + \frac{R_1}{R_2} \right)$$

V_{BB} should be sufficiently large to cover the dead band where both transistors would be off without the bias; moreover, a typical V_{BB} is around 1 V. Also, the output transistors Q_N and Q_P are typically power BJTs, which have monstrous current and voltage ratings and can dissipate a large amount of heat efficiently. With the basic three stages of the audio amplifier investigated, the physical audio amplifier is ready to be designed.

IV. DESIGN PROCEDURE

The three elementary stages of the audio amplifier as discussed in the previous section must be carefully designed with respect to each other in order for the entire amplifier to be stable and to function correctly. First, the input stage must be designed with the theory of the differential amplifier as shown in Figure 4. Unlike the differential pair shown in Figure 4, the output will be taken off of a single leg of the amplifier. Because of this single-ended operation, the governing gain equation for the stage is one half of the original gain as shown below [3].

$$A_d = \frac{\alpha R_C}{2(r_e + R_E)}$$

R_C is the net resistance seen by the collector, R_E is the external resistance seen by the emitter, r_e is the intrinsic emitter resistance, and α is a parameter of the BJT technology. For this design, the common PNP transistor, 2N5401 from ON Semiconductor was used. A common β value of 100 is chosen to represent the chosen transistor, which gives way to an α of as shown below.

$$\alpha = \frac{\beta}{\beta + 1} = \frac{100}{100 + 1} = \frac{100}{101} = 0.99$$

Next, the values of R_C and R_E are chosen to simplify the design process. R_C or the load resistance of the differential pair is set to 1 k Ω while the external emitter resistance, R_E is set to 0 Ω . With most of the variables set, the differential gain, A_d may be found with the discovery of the remaining resistance, r_e . The intrinsic emitter resistance is governed by the equation below [1].

$$r_e = 25 \Omega \left(\frac{1 \text{ mA}}{I_C} \right)$$

With the collector current, I_C , of each leg of the differential pair set to 1 mA for simplicity, the intrinsic emitter resistance is approximately 25 Ω .

Table 1. Designed Parameters of the Input Stage

Variable	Symbol	Value	Units
Alpha Parameter	α	0.99	-
Beta Parameter	β	100	-
Collector Resistance	R_C	1000	Ω
External Emitter Resistance	R_E	0	Ω
Intrinsic Emitter Resistance	r_e	25	Ω
Collector Current	I_C	1	mA
Differential Gain	A_d	19.8	V/V

The gain of the input stage may now be calculated with the designed variables listed above in Table 1.

$$A_d = \frac{\alpha R_C}{2(r_e + R_E)} = \frac{0.99 * 1000}{2(25 + 0)} = \frac{990}{50} = 19.8 \text{ V/V}$$

The true differential gain will be slightly less than the one calculated above because the actual resistance seen by the collector is the parallel combination of R_C and the input resistance of the next stage: the basic gain cell.

The total gain of the audio amplifier is the product of the gains of each stage divided by the negative feedback into the differential amplifier. A total gain of 1000 V/V or 60 dB was chosen for the audio amplifier with a negative feedback attenuation factor of 20 V/V. With these values designed, the gain of the second stage, the common-emitter topology, can be calculated. Considering that the output stage's gain is ideally equal to 1 V/V, the remaining gain of the amplifier must come from the basic gain cell.

Table 2. Designed Parameters of the Voltage Amplification Stage

Variable	Symbol	Value	Units
Alpha Parameter	α	0.99	-
Beta Parameter	β	100	-
Collector Resistance	R_C	24,500	Ω
External Emitter Resistance	R_E	22	Ω
Intrinsic Emitter Resistance	r_e	2.5	Ω
Collector Current	I_C	10	mA
Differential Gain	A_d	1000	V/V
Intrinsic Output Resistance	r_o	135,000	Ω
External Output Resistance	R_{out}	30,000	Ω

An open-loop gain of 20,000 V/V can easily be calculated from the closed-loop attenuation factor of 20 V/V and the desired closed-loop gain of 1000 V/V. The gain of the first stage was approximately 20 V/V, which clearly leaves 1000 V/V to be attained by the second stage.

The second stage is responsible for the high voltage amplification of the audio amplifier. This design calls for 1000 V/V in the common-emitter stage, which is governed by the following gain equation [3].

$$|A_v| = \frac{g_m R_C}{g_m(R_E + r_e) + 1} \cong \frac{R_C}{R_E + r_e} \text{ with } g_m(R_E + r_e) \gg 1$$

A_v is the voltage gain, g_m is the transconductance, R_C is the output resistance seen by the collector of the transistor, and R_E is the external emitter resistance. The collector current of the second stage is set to a value of 10 mA, which derives an intrinsic emitter voltage of 2.5 Ω . This very small value of r_e negatively affects the input stage's gain by sharply reducing the resistance

seen by R_C of the differential pair. For this reason, an external emitter resistance of 22Ω is added to the common-emitter stage to restore the total emitter resistance to 24.5Ω as in the input stage.

The total output resistance as seen by the collector of the transistor must be calculated in order to find the voltage gain. This output resistance will be the parallel combination of intrinsic output resistance of the BJT and the input resistance of the output stage. According to Bob Cordell, “If the driver and output transistor current gains are assumed to be 100 and 50 respectively, the buffering factor will be the product of these, or about 5000” [1]. This assumption is employed and multiplied by the output resistance of the final stage, which is the impedance of the speaker. A 6Ω speaker will be used in this project; therefore, an external output resistance of $30 \text{ k}\Omega$ is seen by the collector of the common-emitter’s BJT. This output resistance will be in parallel with the intrinsic output resistance, which is calculated from the following equation [3].

$$r_o = \frac{V_A + V_{ce}}{I_C}$$

V_A is the Early voltage, V_{ce} is the collector-emitter voltage, and I_C is the collector current. The collector current was previously set to 10 mA , the common 2N5551 NPN BJT has an Early voltage of 100 V as found in the data sheet, and the max collector-emitter voltage is about 35 V from the source. This yields an intrinsic output resistance of $135 \text{ k}\Omega$. The parallel combination of the input resistance of the output stage and the intrinsic output resistance of the common-emitter stage is $24.5 \text{ k}\Omega$. Therefore, the voltage gain of the second stage is the desired value of 1000 V/V as exhibited in the calculation below.

$$|A_v| = \frac{R_C}{R_E + r_e} = \frac{24,500 \Omega}{22 \Omega + 2.5 \Omega} = 1000 \text{ V/V}$$

Finally, the third stage must be developed to complete the basic audio amplifier. The class AB output stage is a standard form that does not require strenuous design; however, the biasing for the output stage, the V_{BE} multiplier, must be designed to sufficiently cover the dead band. As discussed earlier, the dead band is the region in which both of the output transistors are off for the reason that the base-emitter voltages are less than the minimum apparent threshold voltage. Typically the apparent threshold voltage in a BJT is about 0.5 V [3]. The V_{BE} must cover the threshold voltage for both the Q_N and Q_P transistors; therefore, the value of 0.5 V must be doubled to 1 V to provide bias for both base-emitter voltages. Using the equation discussed in Theoretical Background, R_1 and R_2 can be designed as follows. First, let R_1 equal 1 k Ω , and increase the required V_{BB} or bias voltage to 1.5 V to ensure the dead band is completely covered.

$$V_{BB} = V_{BE1} \left(1 + \frac{R_1}{R_2} \right) \Rightarrow R_2 = \frac{R_1}{\left(\frac{V_{BB}}{V_{BE1}} - 1 \right)} = \frac{1000 \Omega}{\left(\frac{1.5 \text{ V}}{0.5 \text{ V}} - 1 \right)} = \frac{1000 \Omega}{2} = 500 \Omega$$

Therefore, the values of the resistors in the V_{BE} multiplier are 1 k Ω for the collector-base resistor, and 500 Ω for the base-emitter resistor. The three basic stages of the audio amplifier have been designed; nevertheless, there are many more additions that are necessary in order to provide a highly functional and stable operation.

Three additional circuit building blocks must be added to the amplifier in order to generate a functional design. These necessities include current sources for the three stages of the amplifier, an input conditioning and filtering circuit, and an output stabilization network known as a Zobel network. First, a current source is required to power each of the three basic stages of the amplifier that were previously designed. As the methods for designing current sources are relatively elementary, only one design procedure will be examined in this document.

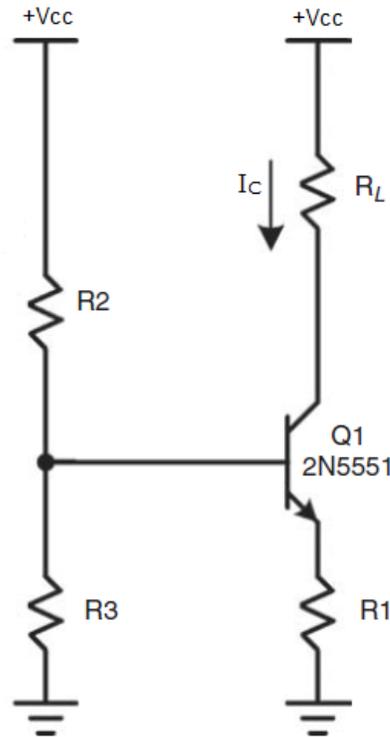


Figure 6. Simple Resistive Current Source Circuit

Starting with the input stage, the required current for each leg of the differential amplifier was set to 1 mA; therefore, a total current of 2 mA is essential for the operation of the differential pair. A simple resistive current source is chosen for the design and is shown below in Figure 6. The voltage divider made up of R_2 and R_3 sets the voltage at the base of Q_1 , which is decreased by the base-emitter voltage of the transistor by about 0.5 V. This voltage appears at the emitter of Q_1 and with the selection of an appropriate resistance for R_1 , the designer can create any value of current desired. The following equation can be derived from the observations made about the simple resistive current source shown in Figure 6.

$$I_C = \frac{\left(\frac{R_3}{R_2 + R_3} \cdot V_{CC}\right) - V_{be}}{R_1}$$

The values for I_C , V_{be} , and V_{CC} are previously known; therefore, in order to design the current source, the values for the resistors R_1 , R_2 , and R_3 must be derived. With a simple observation about the sum of R_2 and R_3 dividing V_{CC} , the values of R_2 and R_3 can be set. The value for R_1 can be easily calculated if the V_{CC} source of 35 V can be easily canceled by a resistance of 35 k Ω . For this reason, R_3 is set to 2 k Ω and R_2 is set to 33 k Ω . Based upon this information, the value of R_1 can be calculated as illustrated below.

$$I_C = \frac{\left(\frac{R_3}{R_2 + R_3} \cdot V_{CC}\right) - V_{be}}{R_1} \Rightarrow R_1 = \frac{\left(\frac{R_3}{R_2 + R_3} \cdot V_{CC}\right) - V_{be}}{I_C}$$

$$= \frac{\frac{2000 \Omega}{33000 \Omega + 2000 \Omega} (35 \text{ V}) - 0.5 \text{ V}}{0.002 \text{ A}} = \frac{(2 \text{ V} - 0.5 \text{ V})}{0.002 \text{ A}} = 750 \Omega$$

Therefore a value of $R_1 = 750 \Omega$ was designed for the simple resistive current source. A similar procedure was conducted for designing the current source for the basic gain cell.

Subsequently, the input conditioning and filtering circuit must be added to the amplifier just before the differential amplifier. Figure 7 shows the input conditioning circuit example in Bob Cordell's *Designing Audio Power Amplifiers* [1]. In this example, the $R_{in}C_{in}$ network is a first-order low pass filter, which is set to a crossover frequency at approximately 250 kHz. This filter is used to block undesirable RF signals.

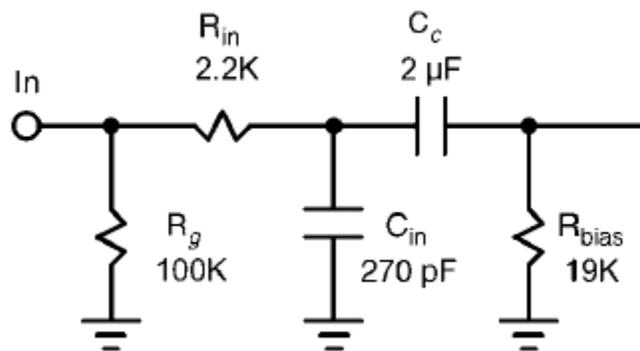


Figure 7. Input Conditioning Circuit

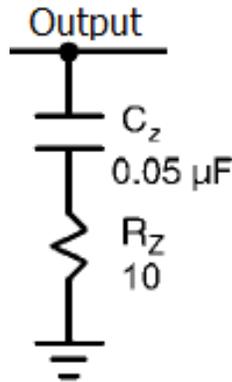


Figure 8. Zobel Network

The resistor, R_g , is a pull-down resistor to keep the input terminal from floating, and R_{bias} exists to keep the DC offset of the input stage at 0 V. The removal of the offset prevents a large DC gain in the basic gain cell, and therefore averts a large unused DC current at the output. The coupling capacitor, C_c , is implemented with a large, exceptional quality film or polypropylene capacitor that blocks DC voltages from entering the amplifier from the source.

The final addition to the amplifier is an output stability network known as the Zobel Network. This snubber circuit as shown in Figure 8 is placed on the output of the amplifier in parallel with the speaker in order to allow a path for the elimination of unwanted high frequencies. These high frequencies are capable of causing instability in the amplifier with certain load conditions for instance at no-load or at highly inductive speaker loads. With the addition of the current sources, the input conditioning circuit, and the Zobel network, the amplifier is complete and is ready to be simulated and tested. Figure 9 displays the complete audio amplifier schematic in its entirety. Resistor and capacitor values have been altered from the exact calculated values to commonly available values.

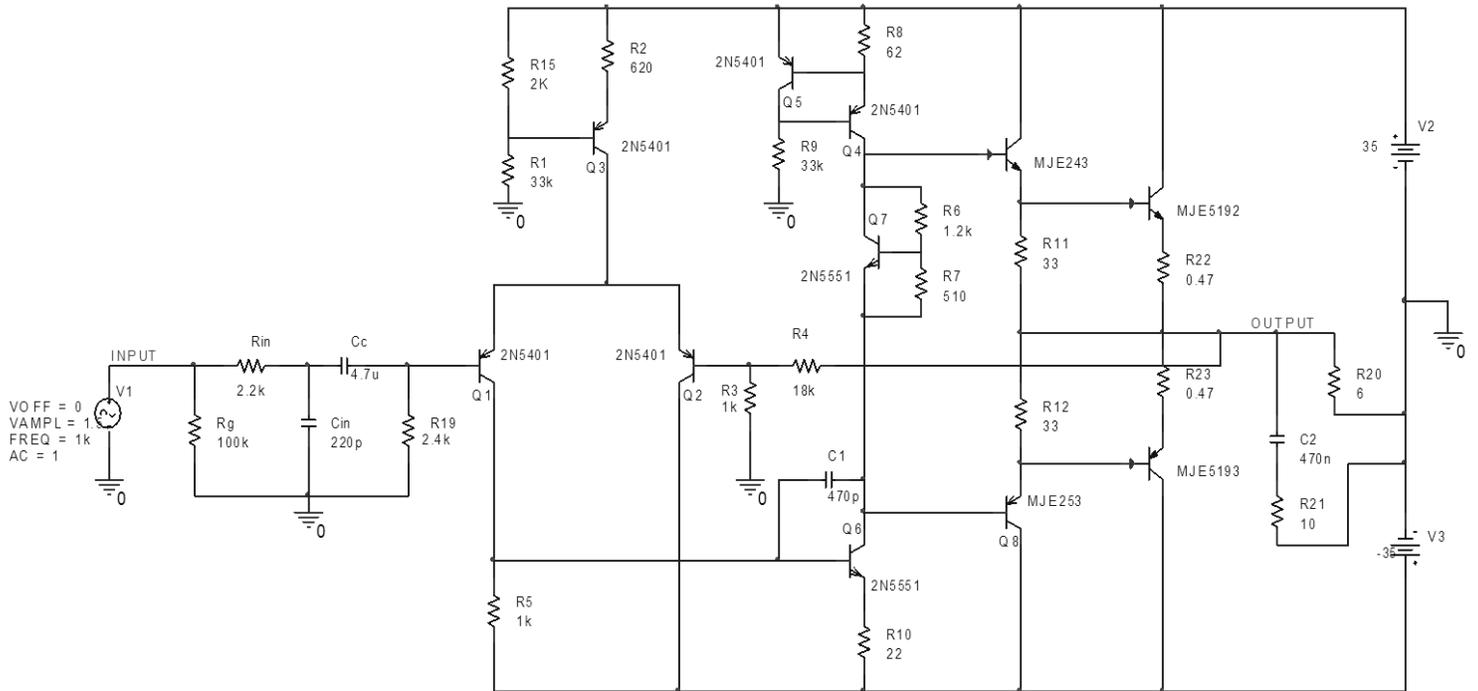


Figure 9. Audio Amplifier Circuit Schematic

The pulse-width modulation circuit was designed with the assistance of the Texas Instruments datasheet on the 555 precision timer [4]. Upon the addition of a clock signal to the trigger input, the circuit shown in Figure 10 will function as a complete pulse-width modulator. The clock input can be implemented with some additional circuitry using an additional 555 timer. The 555 timer that generates the square wave clock signal is operating in what is known as the astable mode of operation. The basics of astable mode are outlined in John Hewes' article, *555 and 556 Timer Circuits*, where he writes, "An astable circuit produces a 'square wave', this is a digital waveform with sharp transitions between low (0V) and high (+Vs). Note that the durations of the low and high states may be different. The circuit is called an astable because it is not stable in any state: the output is continually changing between 'low' and 'high'" [5].

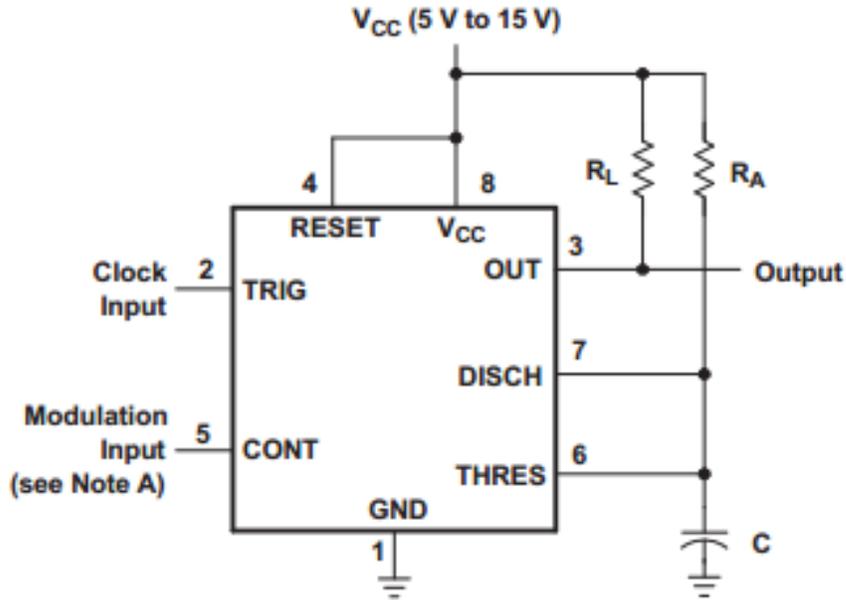


Figure 10. Pulse-Width Modulation with 555 Precision Timer

The following equation governs the operation of the 555 timer in astable mode.

$$f = \frac{1}{\ln(2) \cdot (R_1 + 2R_2) \cdot C_1}$$

The frequency of the clock signal generator will directly translate to the frequency of the triangular carrier signal of the pulse-width modulation circuit. In order to select a carrier frequency, a compromise must be met between the most accurate reproduction of the audio signal (higher frequency) and the limited switching ability of the laser and receiver (lower frequency). A carrier signal of 88.2 kHz was found to be the highest, common sampling frequency for which the receiver could switch. With the frequency selected, the astable clock signal generator can be designed as shown below. R_1 and C_2 are set to 1 k Ω and 10 nF respectively for simplicity.

$$f = \frac{1}{\ln(2) \cdot (R_1 + 2R_2) \cdot C_1} = 88,200 \text{ Hz} = \frac{1}{\ln(2) \cdot (1000 \Omega + 2R_2) \cdot 10 \cdot 10^{-9}}$$

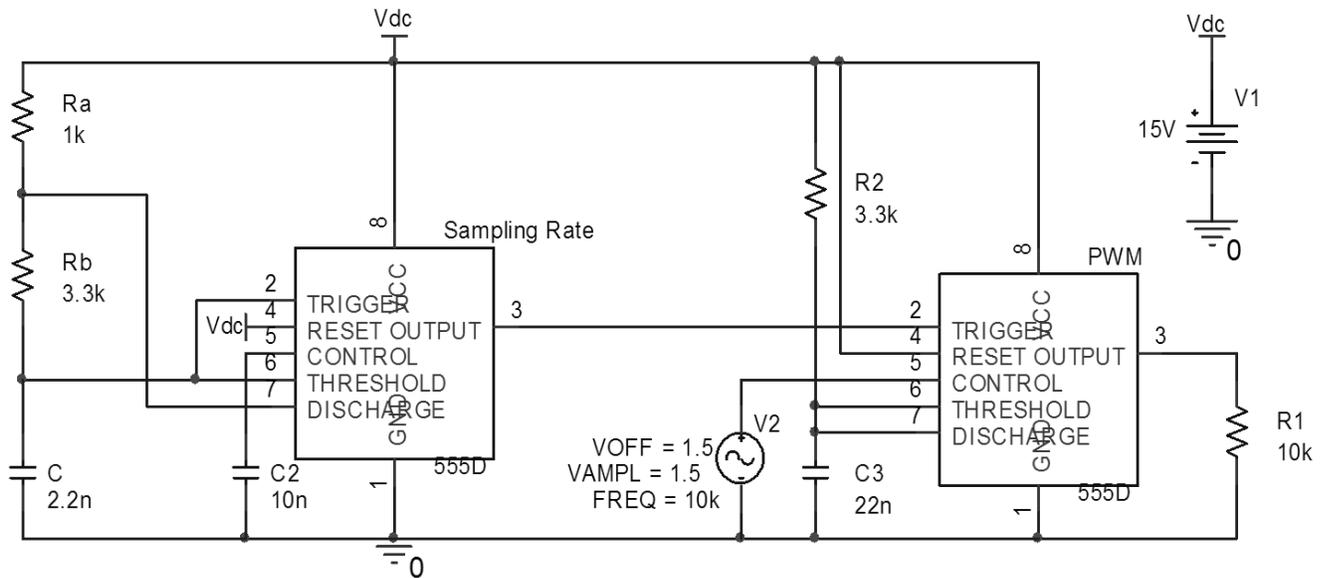


Figure 11. Pulse-width Modulator Circuit Schematic

The value for R_2 would ideally equal $3.2 \text{ k}\Omega$ from the equation above, but a $3.3 \text{ k}\Omega$ resistor will be used as $3.2 \text{ k}\Omega$ is not a standard resistor value. The astable clock signal generator circuit design is complete and is equipped to the input of the pulse-width modulator found in Figure 10.

Commonly known, the 556 timer is an IC that contains two 555 precision timers in a single DIP. This allows the complete pulse-width modulator circuit shown in Figure 11 to be more efficiently implemented with a single 556 timer rather than two separate 555 timers.

V. TESTING AND RESULTS

The designed audio amplifier was evaluated in simulations using the PSPICE circuit simulation software before it was physically implemented on a printed circuit board. Three different types of simulations were conducted on the audio amplifier: a bias point calculation, a transient analysis, and a frequency response plot. The first simulation, the bias point, calculated each individual voltage and current at every node and element in the schematic. The currents of

interest from the design procedure are shown below in Figure 12. As calculated below, the error in the current sources for the differential amplifier and for the common-emitter stages is reasonably low.

$$\%_{Error} = \frac{2.132 \text{ mA} - 2.0 \text{ mA}}{2.0 \text{ mA}} \cdot 100\% = 6.6\%$$

$$\%_{Error} = \frac{10.14 \text{ mA} - 10.0 \text{ mA}}{10.0 \text{ mA}} \cdot 100\% = 1.4\%$$

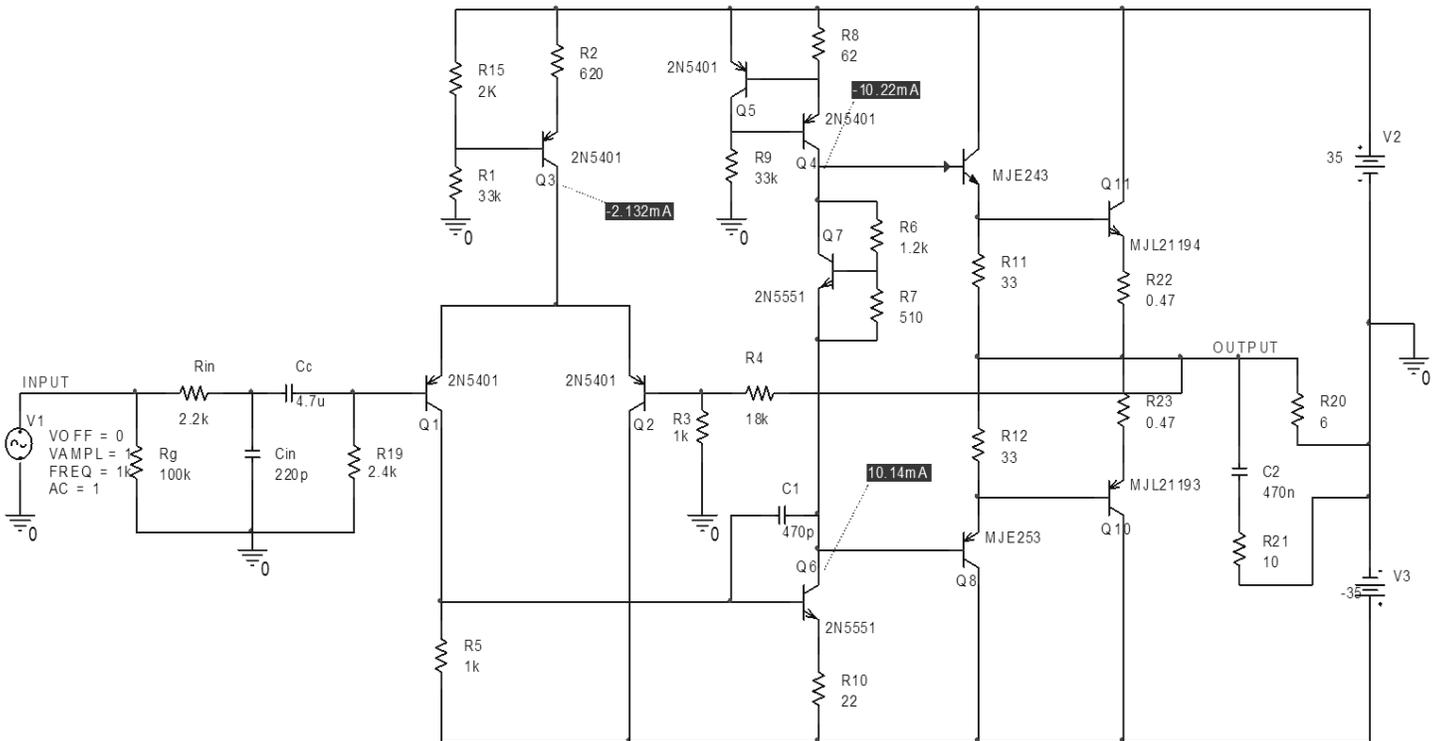


Figure 12. Bias Point Simulation

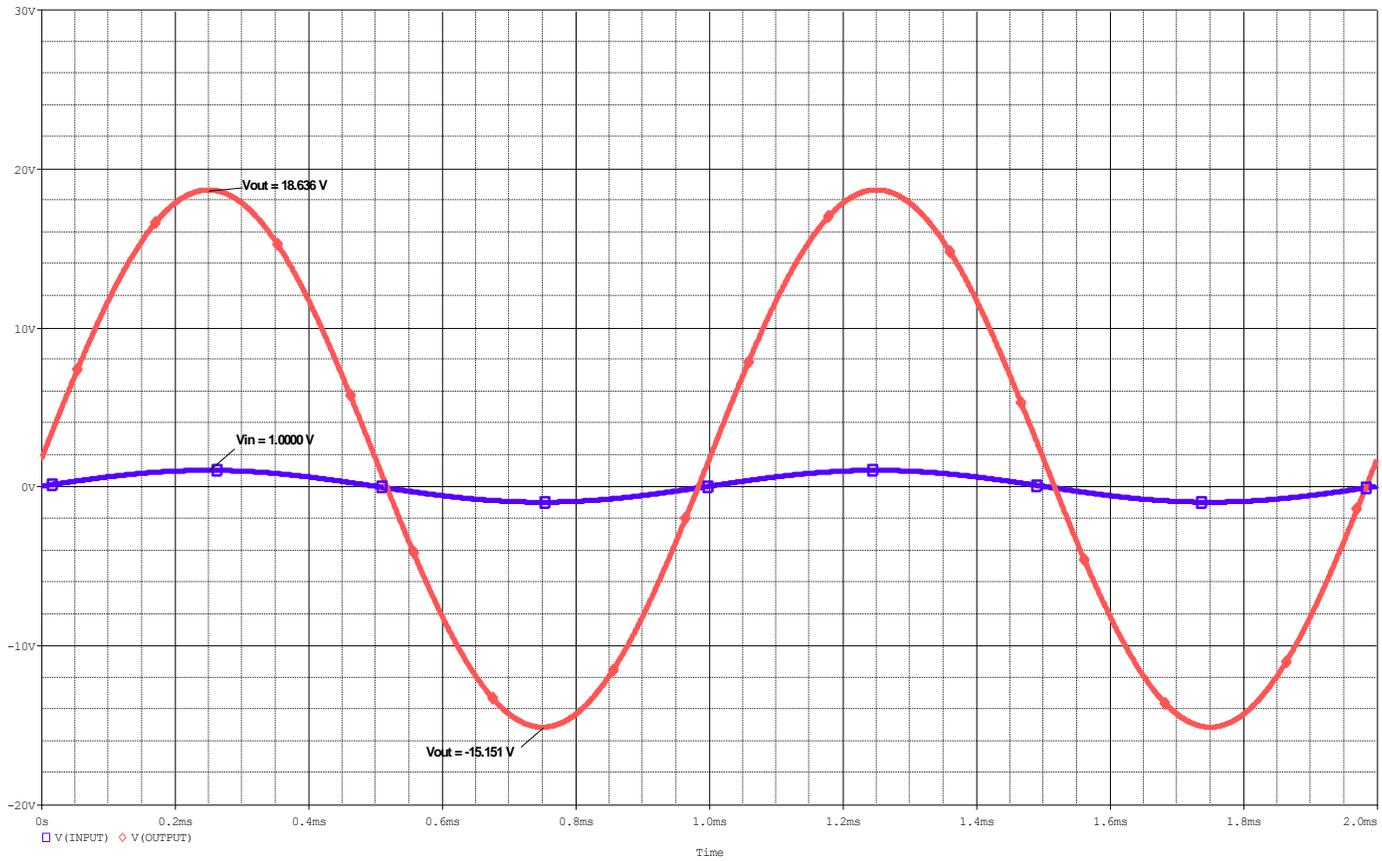


Figure 13. Transient Response of Audio Amplifier

Next, a transient analysis was conducted on the audio amplifier as shown in Figure 13. A clear voltage amplification is taking place as the output waveform is many times larger than the input waveform. By the addition of the conditioning circuit, the gain of the amplifier has been reduced to a more manageable 25 dB. Using the voltage amplitude points found from the transient analysis, the gain was calculated as shown below.

$$A_v(\text{dB}) = 20 \log \left(\frac{v_{out}}{v_{in}} \right) = 20 \log \left(\frac{(18.636 \text{ V} - 15.151)/2}{1 \text{ V}} \right) = 24.55 \text{ dB}$$

This gain results in an error percentage of 1.78%, which is relatively low.

$$\%_{Error} = \frac{25 \text{ dB} - 24.55 \text{ dB}}{25 \text{ dB}} \cdot 100\% = 1.78\%$$

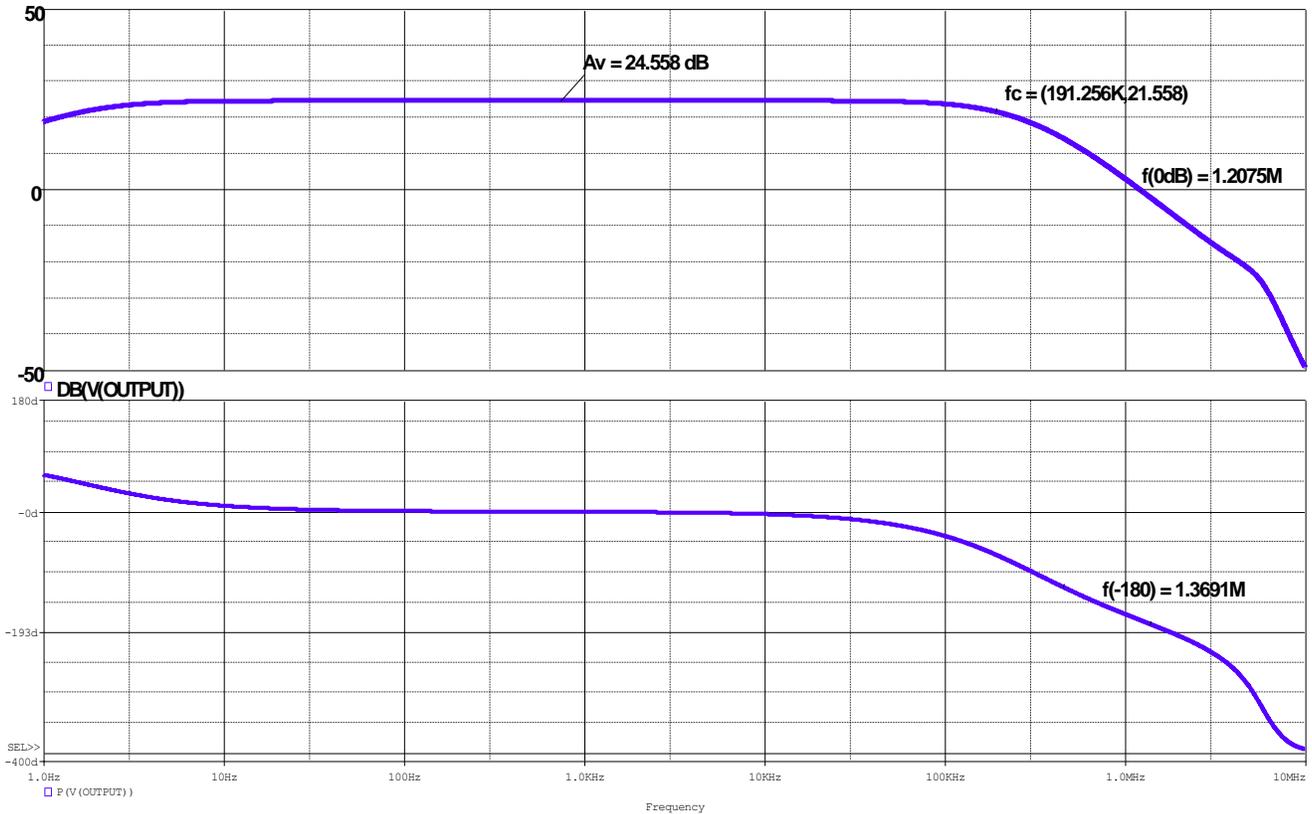


Figure 14. Frequency Response of Audio Amplifier

Finally, a frequency response plot was produced for the audio amplifier as shown in Figure 14. The figure shows many useful data points on the plot including the voltage gain, the cutoff frequency, the 0 dB frequency, and the -180° frequency. These data points can be used to determine the amplifier's stability characteristic. As the anti-phase or -180° frequency is passed the 0 dB frequency the amplifier is stable and has a gain margin of approximately 2 dB and a phase margin of approximately 5° . This very small phase margin can be widened by increasing the miller capacitance of the common-emitter stage; however, the capacitor must be kept relatively small in order to prevent the overall slew-rate of the amplifier from dipping to a dangerously low level [1]. Now that the amplifier design has been confirmed, the schematic can be physically constructed on a printed circuit board.

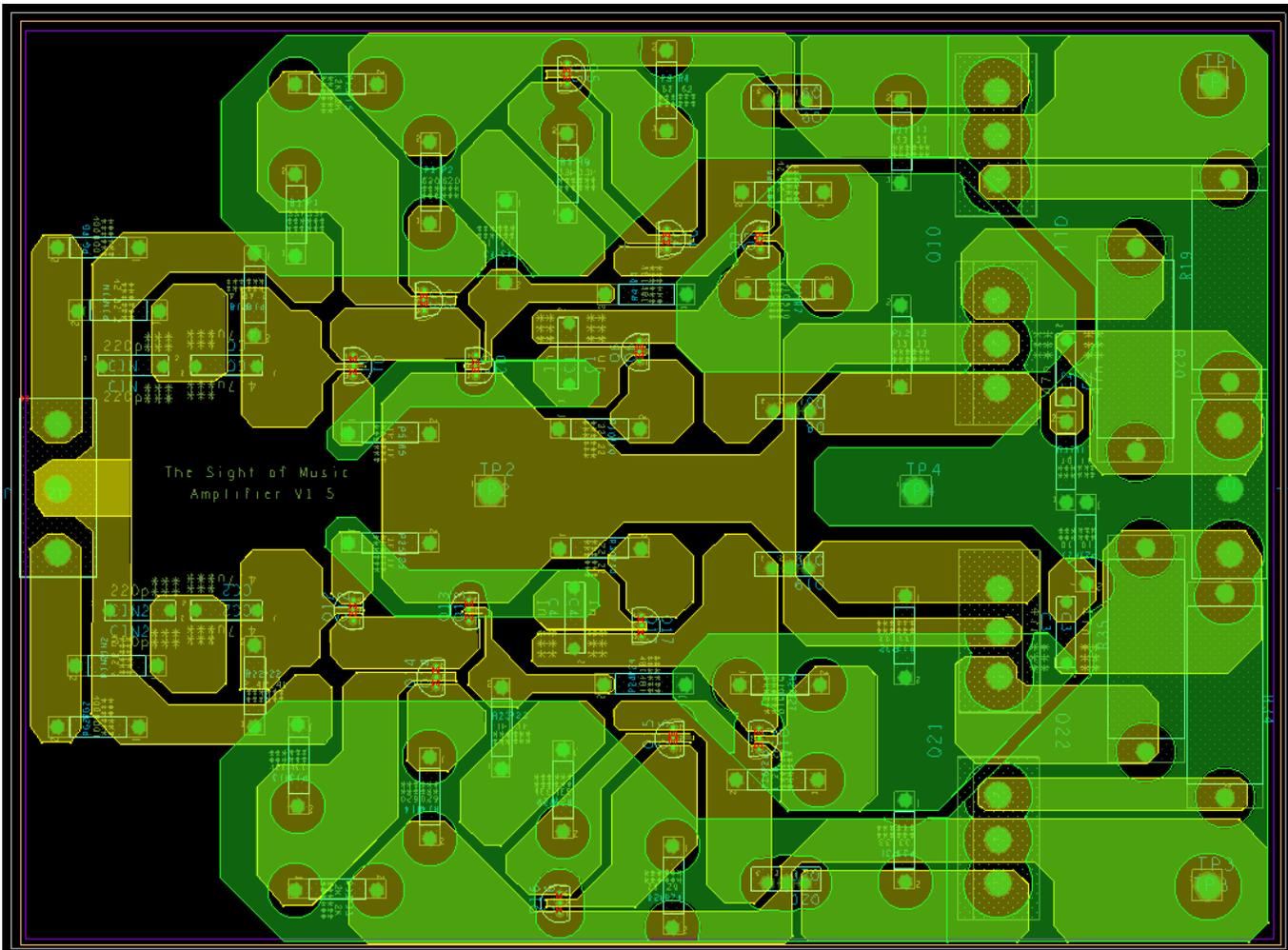


Figure 15. PCB Layout of Audio Amplifier Circuit

Following the design completion and confirmation, the audio amplifier can now be constructed on a PCB. The PCB layout shown in Figure 15 was designed for two audio amplifier circuits in order to have stereo, or two channel, audio signals. This PCB layout implements thick traces to accommodate large currents and minimize the parasitic inductances created by long, thin traces. The majority of the traces are found on the bottom of the board, while two main traces make up the top of the board. The largest trace on the top of the board is a ground plane connecting every ground point in the schematic in one giant trace. The other trace on the top of the board is the negative feedback from the output to the differential amplifier.

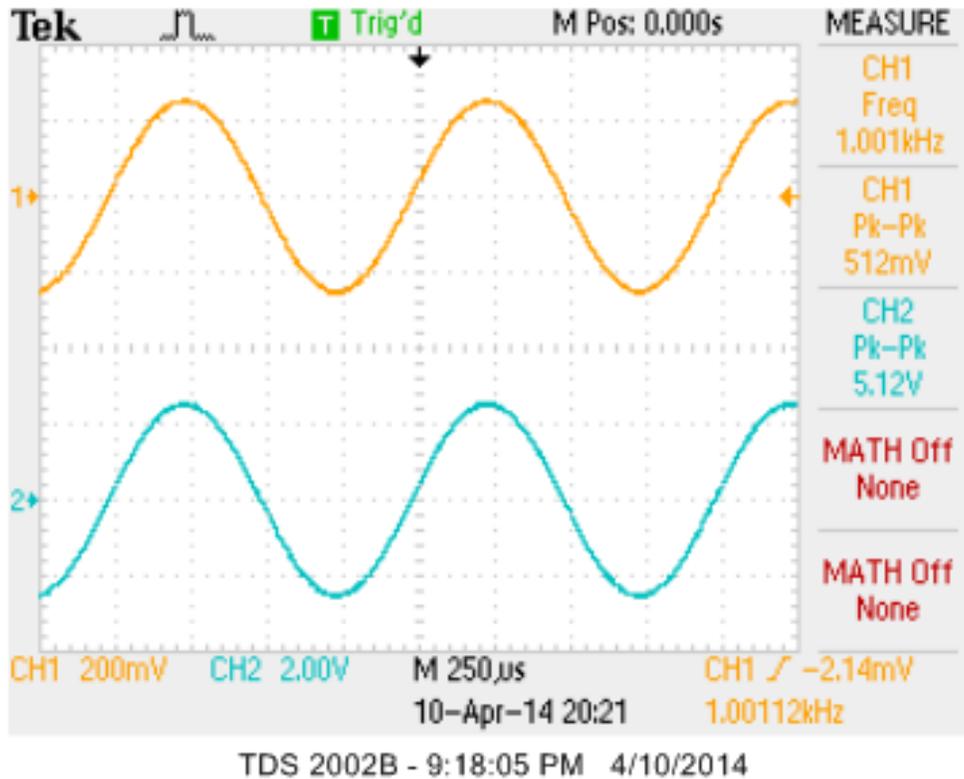


Figure 16. Input and Output Waveforms of the Audio Amplifier

After the PCB had been fabricated using a milling machine, the components were soldered to the board and the amplifier was prepared for testing. The audio amplifier was tested using several tones at various amplitudes in order to ensure its reliability and consistency. The test shown in Figure 16 was conducted at 1 kHz with an amplitude of 0.5 V. The top plot of the oscilloscope measures the input signal to the amplifier, and the bottom plot represents the voltage at the output of the amplifier. From the simple calculations shown below, the amplifier is producing a gain of 10 V/V or 20 dB.

$$A_v = \frac{v_{out}}{v_{in}} = \frac{0.512 \text{ V}}{5.12 \text{ V}} = 10 \text{ V/V} \Rightarrow A_v(dB) = 20 \log\left(10 \frac{\text{V}}{\text{V}}\right) = 20 \text{ dB}$$

The error percentage of the physically tested gain is 20%, which is substantially higher than the calculated and simulated gains.

$$\%Error = \frac{25 \text{ dB} - 20 \text{ dB}}{25 \text{ dB}} \cdot 100\% = 20\%$$

This larger than expected error percentage is most likely due to the non-idealities of the real world including the unwanted finite resistance, inductance, and capacitance of the PCB as well as the components that make up the amplifier. Also, as shown in the Design Section, the gain is dependent on the load; therefore, a change in the load impedance may reduce the overall gain. The long wires that attach the speakers to the amplifier may increase the overall impedance of the load, and are not taken into account during the calculation or simulation processes.

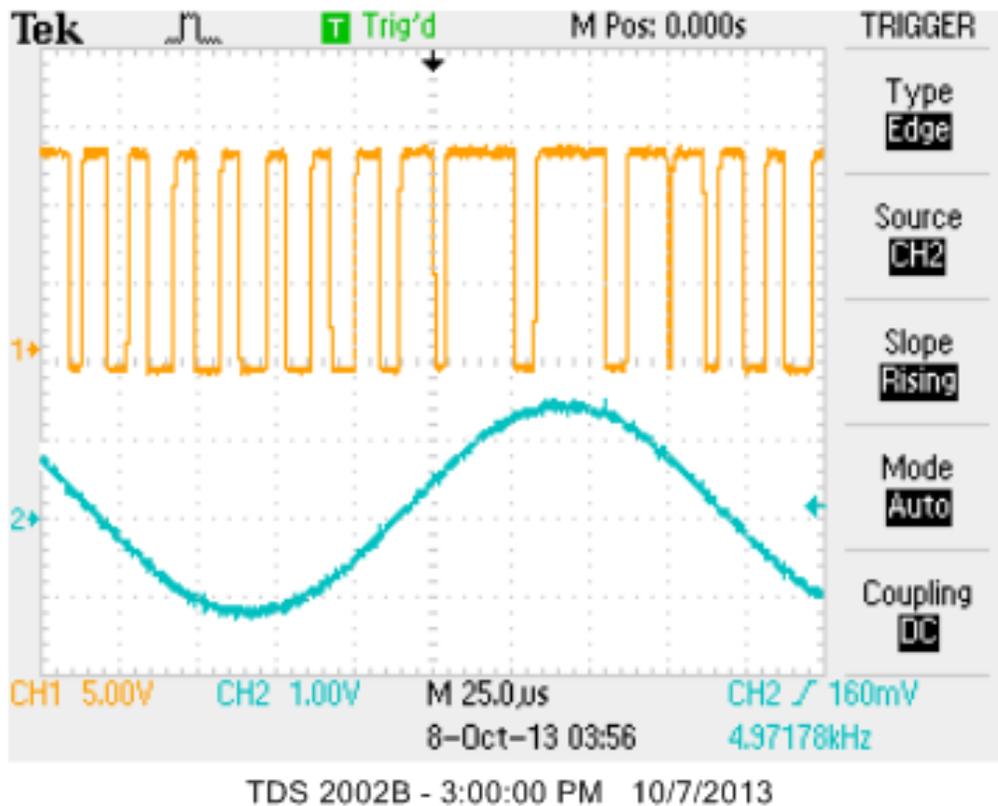


Figure 17. Pulse-width Modulation Waveform

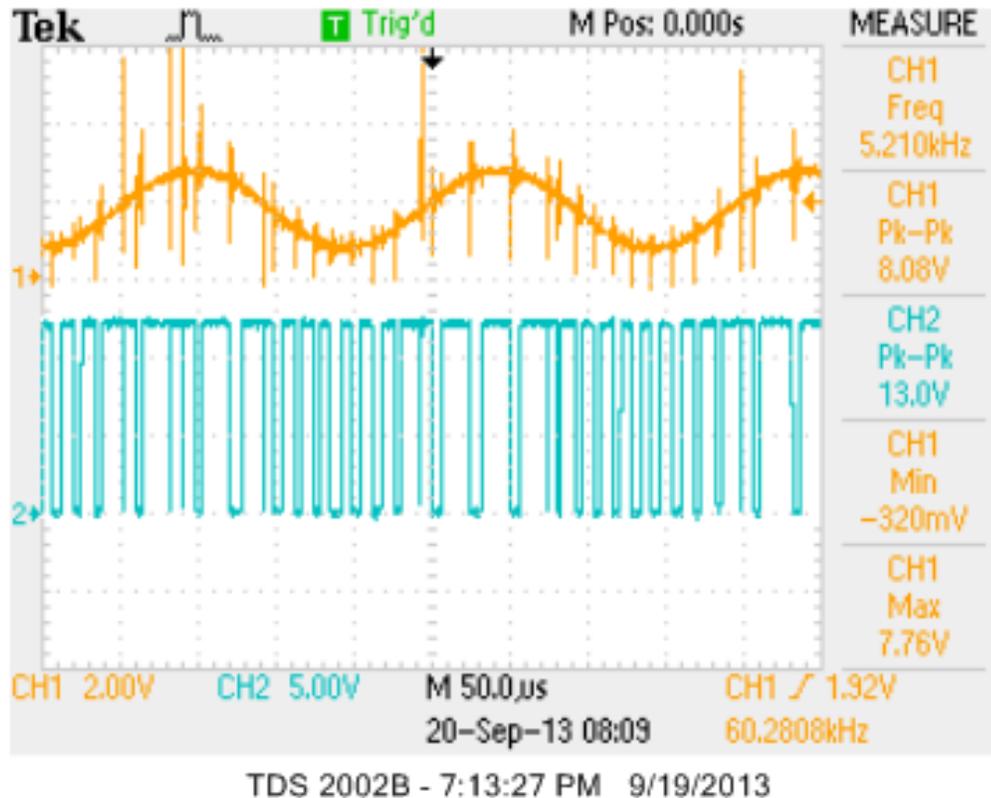


Figure 18. Rectification of Sinusoidal Signal from PWM Signal

The transmission system was tested for functionality in two stages. The first stage included a test to confirm the pulse-width modulation circuit's ability to modulate a sinusoidal input signal. The second stage's test consisted of the pulse-width demodulation circuit's ability to rectify the PWM signal back to the original sinusoid after transmission.

A 5 kHz sinusoidal signal was used in the testing process. Figure 17 illustrates the first stage of testing: the pulse-width modulation occurring on a 5 kHz sinusoid. The pulse widths are greater when the amplitude of the sinusoid is positive, and the pulse widths are much less when the sinusoid is negative. This observation confirms the functionality of the PWM circuit. For the second test, the PWM signal is passed through a low pass filter, which ideally removes the carrier frequency and leaves only the original sinusoid. As no filter is ideal, the resulting

sinusoid resembles the top plot in Figure 18. The fundamental 5 kHz frequency is clearly evident, but harmonics of the carrier signal also exist on the output sinusoid. The system has successfully transmitted a sinusoidal signal from a frequency generator to the speaker at the output and is ready to transmit audio.

VI. CONCLUSION

An alternative method for transmitting audio signals via the use of laser technology was presented in this document. Through the design procedure using theoretical calculations, simulations using PSPICE computer software, and examination of the finished product with an oscilloscope, the audio amplifier was greatly investigated and verified for desired functionality. In addition, the pulse-width modulation technique was examined and implemented with precision 555 timers in order to transmit audio signals via laser. The main weakness of the laser transmission method presented here is the requirement for the laser's beam to be directly aimed at the receiver. Even with this impediment, the pulse-width modulation laser system may be regarded as a defensible method of audio transmission, and despite its drawbacks, it is capable of producing accurate sound reproduction.

VII. REFERENCES

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